DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on March 4, 2008 has been entered.

Response to Arguments

2. Applicant's arguments filed March 4, 2008 have been fully considered but they are not persuasive.

3. In response to applicant's arguments (Remarks page 7 and 10), the recitation “a method for preprocessing audio data to be processed by a codec having a variable coding rate” has not been given patentable weight because the recitation occurs in the preamble. A preamble is generally not accorded any patentable weight where it merely recites the purpose of a process or the intended use of a structure, and where the body of the claim does not depend on the preamble for completeness but, instead, the process steps or structural limitations are able to stand alone. See *In re Hirao*, 535 F.2d 67, 190 USPQ 15 (CCPA 1976) and *Kropa v. Robie*, 187 F.2d 150, 152, 88 USPQ 478, 481 (CCPA 1951).

4. Applicant also argues that, "Nowhere in Malvar is there any disclosure or suggestion regarding the use of automatic gain control to adjust the amplitude of audio data of the decided interval, which is to be encoded in a low bit rate in the codec, before the audio data is processed by the codec, as claimed." (Remarks page 8), adding that, "In column 2 lines 41-56, Malvar merely describes automatic gain control as an example of enhancement operators incurring a processing delay that will be added to the codec delay, without mentioning its detailed functions"; however the examiner respectfully disagrees. ***DeJaco*** determines an interval of audio data to be encoded in a low bit rate in the codec (column 3 lines 56-65). Additionally, ***Malvar*** states, "For instance, audio paths used with current codecs may include, *prior to processing by the codecs (emphasis added)*, a signal enhancement module." (column 2 lines 42-44). ***Malvar*** then continues with an example, concluding that, "Other enhancement operators may include automatic gain control noise reducers, etc." (column 2 lines 49-50). Automatic gain control is a common method used to adjust the amplitude of an audio signal prior to further processing. Therefore ***Malvar*** teaches the use of automatic gain control to adjust the amplitude of audio data as claimed, and ***DeJaco*** and ***Malvar*** combined disclose "adjusting the amplitude of audio data of the decided interval before the audio data is processed by the codec", as recited in claim 4.

5. Applicant also argues that ***Malvar*** does not disclose adjusting the amplitude, as recited in claim 4, since "the enhancement operators are required in the codec, not for adjusting the amplitude of the audio data such that the audio data is coded in a higher rate by the codec but for processing degraded spee[ch] in the codec"; however this

argument is considered moot. The limitation upon which applicant relies, i.e. adjusting the amplitude of the audio data such that the audio data is coded in a higher rate by the codec, is a recitation of intended use. A recitation of the intended use of the claimed invention must result in a structural difference between the claimed invention and the prior art in order to patentably distinguish the claimed invention from the prior art. If the prior art structure is capable of performing the intended use, then it meets the claim.

6. In response to applicant's argument that there is no suggestion to combine the references, the examiner recognizes that obviousness can only be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some teaching, suggestion, or motivation to do so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art. See *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988) and *In re Jones*, 958 F.2d 347, 21 USPQ2d 1941 (Fed. Cir. 1992). In this case, ***Malvar*** discloses automatic gain control as one of many possible enhancement functions used prior to processing by the codec. Therefore it would have been obvious to use the known method of automatic gain control in ***DeJaco***, in order to improve the system in a predictable way.

7. Applicant also argues that, "Malvar, however, does not disclose performing AGC preprocessing of frames of audio data classified as based on a characteristic of audio data before the audio data is processed by a predetermined codec.", however the examiner respectfully disagrees. ***DeJaco*** discloses classifying the audio data selected based on a characteristic of the audio data (column 2 lines 15-18 and lines 39-42).

Additionally, ***Malvar*** states, "For instance, audio paths used with current codecs may include, *prior to processing by the codecs (emphasis added)*, a signal enhancement module." (column 2 lines 42-44). ***Malvar*** then continues with an example, concluding that, "Other enhancement operators may include automatic gain control noise reducers, etc." (column 2 lines 49-50). Automatic gain control is a common method used to adjust the amplitude of an audio signal prior to further processing. Therefore ***Malvar*** teaches the use of automatic gain control to adjust the amplitude of audio data as claimed, and ***DeJaco*** and ***Malvar*** combined disclose "performing AGC preprocessing on frames of audio data classified based on a characteristic of audio data before the audio data is processed by a predetermined codec" as claimed in claim 2.

8. Applicant also argues that, "Nowhere in Malvar if there any disclosure or suggestion regarding performing AGC processing of al frames of audio data in case the audio data includes monophonic sound or performing AGC processing of selected frames in case the audio data includes polyphonic sound, as claimed" (Remarks page 11). The examiner agrees, and notes the Final Office Action dated September 4, 2007 used ***Davis***, in combination with ***DeJaco*** and ***Malvar***, to reject this limitation. ***Davis*** performs premphassis, or preprocessing, of a signal based on a ratio of high frequency energy to low frequency energy (column 2 lines 50-67). Monophonic music, having one tone or pitch, would have a constant ratio of high frequency energy to low frequency energy; therefore any preemephasis or preprocessing needed would take place over every frame of the signal. Polyphonic music would have a ratio of high frequency energy to low frequency energy that varies, depending on the tones being played at that time.

This suggests that the amount of preemphasis or preprocessing needed would depend on the characteristics of the signal being played during a particular frame. Therefore, ***DeJaco*** and ***Malvar*** in view of ***Davis*** disclose performing AGC processing of all frames of audio data in case the audio data includes monophonic sound or performing AGC processing of selected frames in case the audio data includes polyphonic sound as recited in claim 2.

9. In response to applicant's argument that there is no suggestion to combine the references, the examiner recognizes that obviousness can only be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some teaching, suggestion, or motivation to do so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art. See *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988)and *In re Jones*, 958 F.2d 347, 21 USPQ2d 1941 (Fed. Cir. 1992). In this case, ***Malvar*** discloses automatic gain control as one of many possible enhancement functions used prior to processing by the codec. In addition, ***Davis*** discloses selective processing of a signal based on the frequency characteristics of that signal. Therefore it would have been obvious to use the known method of automatic gain control and selective processing based on characteristics of the signal in ***DeJaco***, in order to improve the system in a predictable way.

Claim Objections

Claims 2, 9 are objected to because of the following informalities: AGC, in line 4 of claim 2 and line 8 of clam 9, should read, “automatic gain control (AGC)”. Appropriate correction is required.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claims 4 and 6-8 are rejected under 35 U.S.C. 103(a) as being unpatentable over ***DeJaco*** (5,742,734) in view of ***Malvar*** (6,029,126).

10. As per claims 4 and 6, ***DeJaco*** discloses a method for preprocessing audio data to be processed by a codec having variable coding rate, comprising the steps of:

deciding an interval of audio data that is to be encoded in a low bit rate in said codec (column 3 lines 56-65).

DeJaco does not explicitly disclose adjusting the amplitude of audio data of the decided interval before the audio data is processed by the codec, such that the audio data in the interval may be encoded in a bit rate higher than or equal to said low bit rate

when processed by the codec. However, ***DeJaco*** does disclose that previous speech coding systems do not correctly determine when low energy unvoiced speech is input (column 1 lines 40-52). The systems often mistake low energy unvoiced speech as noise and encode the signal at a lower bit rate, causing degradation in speech quality during speech reconstruction (column 1 lines 40-52). ***Malvar*** discloses that signal enhancement functions are often used to enhance a signal prior to processing by the codec, automatic gain control being one of those functions (column 2 lines 41-51). The enhancement functions are used to transform the signal in order to increase encoding accuracy.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use the known enhancement function of adjusting the amplitude of the audio data in ***DeJaco***, since it would improve the system in a similar way, enabling the system to distinguish the unvoiced speech signal from noise then correctly encode the frame at a higher bit rate, thus reducing encoding errors and increasing resolution during speech reconstruction.

11. As per claim 7, ***DeJaco*** discloses a method for preprocessing audio data to be processed by a codec having variable coding rate, wherein the codec is capable of determining whether data fed to the codec is noise signal or not, comprising the steps of:

Deciding whether a frame in the audio data would be determined as noise signal when the audio data is processed by the codec (column 2 lines 15-18 and lines 39-42, *the input signal is analyzed to determine the presence of speech or music. If the input signal is neither speech nor music, then it must be noise or silence with background noise*).

DeJaco does not disclose that if the signal is determined as noise signal, preprocessing the frame such that the preprocessed frame is not determined as noise when processed by the codec. However, ***DeJaco*** does disclose that previous speech coding systems do not correctly determine when low energy unvoiced speech is input (column 1 lines 40-52). The systems often mistake low energy unvoiced speech as noise and encode the signal at a lower bit rate, causing degradation in speech quality during speech reconstruction (column 1 lines 40-52). ***Malvar*** discloses that signal enhancement functions are often used to enhance a signal prior to processing by the codec, automatic gain control being one of those functions (column 2 lines 41-51). The enhancement functions are used to transform the signal in order to increase encoding accuracy.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to preprocess the frame such that the preprocessed frame is not determined as noise when processed by the codec, if the signal is determined to be a noise signal in ***DeJaco***, since it would enable the system to distinguish the unvoiced speech signal from noise then correctly encode the frame at a higher bit rate, thus reducing encoding errors and increasing resolution during speech reconstruction.

12. As per claim 8, ***DeJaco*** does not disclose preprocessing audio data before the audio data is transmitted through the transmission channel, such that the audio data is processed in the codec in a higher bit rate from the bit rate without the preprocessing. However, ***DeJaco*** does disclose the compression of an audio signal prior to transmission (column 1 lines 40-43). ***DeJaco*** also discloses that previous speech coding systems do not correctly determine when low energy unvoiced speech is input (column 1 lines 40-52). The systems often mistake low energy unvoiced speech as noise and encode the signal at a lower bit rate, causing degradation in speech quality during speech reconstruction (column 1 lines 40-52). ***Malvar*** discloses that signal enhancement functions are often used to enhance a signal prior to processing by the codec, automatic gain control being one of those functions (column 2 lines 41-51). The enhancement functions are used to transform the signal in order to increase encoding accuracy.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use the known enhancement function of adjusting the amplitude of the audio data in ***DeJaco***, since it would improve the system in a similar way, enabling the system to distinguish the unvoiced speech signal from noise then correctly encode the frame at a higher bit rate, thus reducing encoding errors and increasing resolution during speech reconstruction.

Claims 2, 3 and 9 are rejected under 35 U.S.C. 103(a) as being unpatentable over ***DeJaco*** in view of ***Malvar,*** further in view of ***Davis*** (4,539,526).

13. As per claim 2, ***DeJaco*** discloses a method for preprocessing audio data to be processed by a predetermined codec having variable coding rate, comprising the steps of:

classifying the audio data based on a characteristic of the audio data (column 2 lines 15-18 and lines 39-42, *the input signal is analyzed to determine the presence of speech or music then processed accordingly*).

DeJaco does not disclose in case the audio data includes monophonic sound, performing AGC (automatic gain control) preprocessing of all frames, and in case the audio data includes polyphonic sound, performing AGC preprocessing of selected frames. ***Malvar*** discloses that signal enhancement functions are used to enhance a signal prior to processing by the codec, automatic gain control being one of those functions (column 2 lines 41-51). The enhancement functions are used to transform the signal in order to increase encoding accuracy. In addition, ***Davis*** discloses a system that performs preemphasis on a signal prior to encoding, the preemphasis based on a ratio of high frequency energy to low frequency energy (column 2 lines 50-67). ***Davis*** also discloses that conventionally, preemphasis is used to adjust a signal level to below a maximum level or above a noise level. Monophonic music, having one tone or pitch, would have a constant ratio of high frequency energy to low frequency energy; therefore any preemephasis or preprocessing needed would take place over every frame of the

signal. Polyphonic music would have a ratio of high frequency energy to low frequency energy that varies, depending on the tones being played at that time. The amount of preemphasis or preprocessing needed would depend on the tones being played during a particular frame.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to adjust the gain for all frames in monophonic music and selected frames in polyphonic music in ***DeJaco***, in order to adjust the signal to below a maximum level and above a minimum noise level, thus reducing errors in a bandlimited application, such as encoding and decoding prior to transmission, as indicated in ***Davis*** (column 1 lines 31-36 and column 2 lines 46-65).

14. As per claim 3, ***DeJaco*** in view of ***Malvar*** and further in view of ***Davis*** disclose a method in accordance with claim 2, but ***DeJaco*** does not explicitly disclose wherein the step of performing AGC preprocessing of selected frames include deciding whether a frame in the audio data includes noise signal or not. However, ***DeJaco*** does disclose determining whether an input signal is noise or not (column 2 lines 15-18 and lines 39-42, *the input signal is analyzed to determine the presence of speech or music as compared to background noise. If the input signal is neither speech nor music, then it must be noise or silence with background noise*). In addition, ***Malvar*** discloses that signal enhancement functions are used to enhance a signal prior to processing by the codec, automatic gain control being one of those functions (column 2 lines 41-51). The

enhancement functions are used to transform the signal in order to increase encoding accuracy.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to determine if an input frame is noise or not during AGC preprocessing in ***DeJaco,*** in order to distinguish the unvoiced speech signal from noise, then correctly encode the frame at a higher bit rate, thus reducing errors and increasing resolution once it is decoded.

15. As per claim 9, ***DeJaco*** discloses an apparatus for preprocessing audio data to be processed by a codec having variable coding rate, the apparatus being apart from the predetermined codec, comprising:

Means for classifying the audio data based on the characteristic of the audio data (column 2 lines 15-18 and lines 39-42, *the input signal is analyzed to determine the presence of speech or music then processed accordingly*);

Means for deciding an interval of the audio data that is to be encoded in a low bit rate in said codec (column 3 lines 56-65).

DeJaco does not disclose deciding an interval of the audio data that is to be encoded in a low bit rate in said codec in case the audio data is determined to include polyphonic sound based on the classification, or means for performing AGC preprocessing of all frames before the audio data is subject to the codec in the case

audio data is determined to include monophonic sound based on the classification, and performing AGC preprocessing of frames of the decided interval before the audio data is subject to the codec in case the audio data is determined to include polyphonic sound based on the classification. However, ***DeJaco*** does disclose that previous speech coding systems do not correctly determine when low energy unvoiced speech is input (column 1 lines 40-52). The systems often mistake low energy unvoiced speech as noise and encode the signal at a lower bit rate, causing degradation in speech quality during speech reconstruction (column 1 lines 40-52). ***Malvar*** discloses that signal enhancement functions are used to enhance a signal prior to processing by the codec, automatic gain control being one of those functions (column 2 lines 41-51). The enhancement functions are used to transform the signal in order to increase encoding accuracy. In addition, ***Davis*** discloses a system that performs preemphasis, or preprocessing, on a signal prior to encoding, the preemphasis based on a ratio of high frequency energy to low frequency energy (column 2 lines 50-67). ***Davis*** also discloses that conventionally preemphasis is used to adjust a signal level to below a maximum level or above a noise level. Monophonic music, having one tone or pitch, would have a constant ratio of high frequency energy to low frequency energy; therefore any preemephasis or preprocessing needed would take place over every frame of the signal. Polyphonic music would have a ratio of high frequency energy to low frequency energy that varies, depending on the tones being played at that time. The amount of preemphasis or preprocessing needed would depend on the tones being played during a particular frame.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to disclose deciding an interval of the audio data that is to be encoded in a low bit rate in said codec in case the audio data is determined to include polyphonic sound based on the classification, and perform AGC preprocessing of all frames before the audio data is subject to the codec in the case audio data is determined to include monophonic sound based on the classification, and perform AGC preprocessing of frames of the decided interval before the audio data is subject to the codec in case the audio data is determined to include polyphonic sound based on the classification in ***DeJaco***, in order to adjust the signal to below a maximum level and above a minimum noise level, thus reducing errors in a bandlimited application, such as encoding and decoding prior to transmission, as indicated in ***Davis*** (column 1 lines 31-36 and column 2 lines 46-65).

Claim 5 is rejected under 35 U.S.C. 103(a) as being unpatentable over ***DeJaco*** in view of ***Malvar*** as applied to claim 4 above, and further in view of ***Forse*** (4,912,766).

16. ***DeJaco*** in view of ***Malvar*** discloses a method in accordance with claim 4, however neither ***DeJaco*** nor ***Melvar*** further disclose wherein the adjusting step comprises the steps of: calculating signal levels of the audio data, deciding smoothed gain coefficients based on signal levels, and generating preprocessed audio data by multiplying the smoothed gain coefficients to the audio data in the decided interval. However, ***Malvar*** discloses that signal enhancement functions are used to enhance a

signal prior to processing by the codec, automatic gain control being one of those functions (column 2 lines 41-51). In addition, ***Forse*** discloses a system that uses automatic gain control in a speech application (column 1 lines 45-58). The system inputs a speech signal, determines spectral parameters, stores gain coefficients for each spectral parameter then uses the lowest of the gain coefficients to adjust the magnitude of the spectral parameters.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to apply the known technique of determining gain coefficients, and multiplying those coefficients by the input signal in ***DeJaco***, since it would improve the system in a similar way, enabling the system to distinguish the unvoiced speech signal from noise then correctly encode the frame at a higher bit rate, thus reducing encoding errors and increasing resolution during speech reconstruction.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Dorothy Sarah Siedler whose telephone number is 571-270-1067. The examiner can normally be reached on Mon-Thur 9:30am-5:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

DSS
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